

Asterisk 1.6.1 on openSUSE

Mohammad Edwin Zakaria

medwin@opensuse.org

Notes:

This article is derived freely from <http://medwinz.blogsome.com>. I use Bahasa Indonesia in explaining the dialplan so I hope everyone will understand it clearly. This is not an academic paper, I'm not an expert, but only a practitioner so it may not suit for academic presentation.

PART 1

I use Asterisk 1.6.1.5 from openSUSE repository. Actually I built a custom 64 bit appliance using KDE 4.3 from factory repositories through SUSE Studio and took Asterisk from openSUSE Build Service repositories. I also included DAHDI (Digium Asterisk Hardware Device Interface), but during the implementation I have a problem with Indonesia PSTN telephone signaling so I should download dahdi trunk version from digium subversion server to make the digium card works.

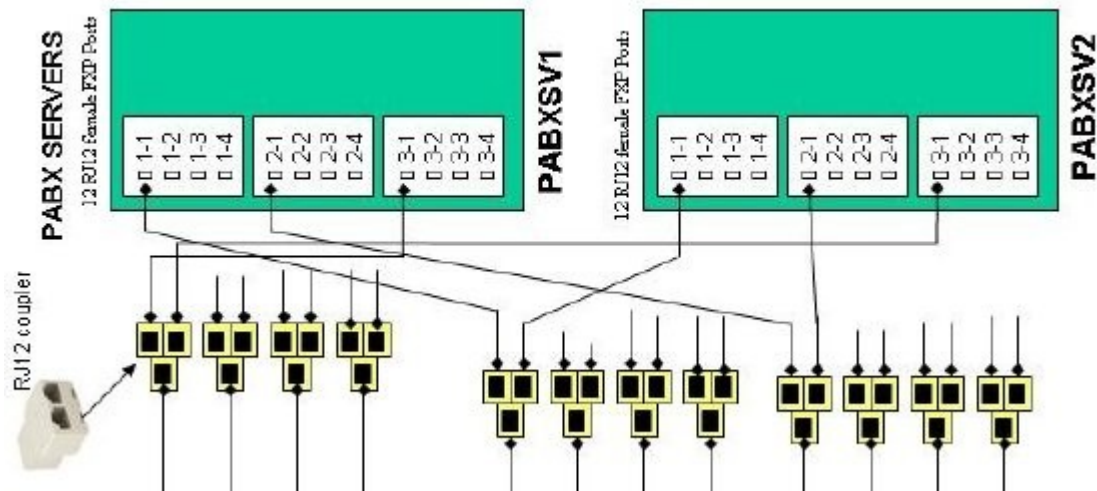
Here are the hardware I use:

1. 2 HP tower based server with 8 GB memory (it is overkill actually, but the owner insist it) running in high availability. See the pictures [here](#) and [here](#).
2. 10 PSTN lines
3. 3 Digium [TDM 410P cards](#) (with 4 FXO ports per card and hardware echo canceler) per server
4. several [RJ12 coupler](#)
5. RJ 12 cables
6. 2 [Zed-3 GS8 GSM gateway](#), each with 2 GSM modules
7. Several [Polycom IP-330](#) with PoE
8. Polycom [KIRK Wireless Server 600V3](#)
9. Several Polycom [DECT 3040 Wireless](#) Handset

Digium and Polycom prices are expensive but the quality of the sound is very good. There are some alternatives for the IP Phone like Grandstream and Aastra that also can be used.

In this project, Asterisk will be use to setup the voip communication between this site in Denpasar/Bali with the headquarter (HQ) in Jakarta as well as with other regional center in Java and Sumatera. Also Asterisk will act as traditional PBX to connect this site to PSTN lines as well as to GSM/CDMA lines. Every conversation through the PABX will be recorded by monitor application in Asterisk.

Before we go any further lets discuss a logical design about our setup. There is one HQ and several remote sites including Bali. These sites is special because it's also act as second node beside HQ that can receive and transmit voip traffic to other center. The setup of every site is similar like the diagram below.



All the digium card provide 12 lines of PSTN, in this case we only use 10 lines. We then use RJ 12 coupler so that every line goes to 2 PBX server, PABXSV1 and PABXSV2. The PABXSV2 will become the backup asterisk in case the PABXSV1 is downed. We use vrrpd to control the service so that PABXSV2 can take over all the service from PABXSV1.

I use stock asterisk and dahdi from OBS. While the asterisk is ok, dahdi in the OBS is not sufficient for Indonesia telephone lines (at least at the time I made the appliance). The root cause of the problem is that Indonesia PSTN line provided by Telkom is already equipped with the CID (caller identifier) but the service is not open to the end customer until the customer pay the service charge. But actually the CID is there and asterisk knows it but cannot open it. So it can answer the ring but if another call comes, suddenly it confuses how to handle it and hangup the line. Off course we should make a good configuration not just downloading the trunk version.

PART 2

In this second part I will explain step-by-step configuration to use [our appliance](#) to build an Asterisk PBX server. Without further ado, here is the list:

1. Install the Digium card on the PCI slot
2. Install our appliance. You can also use any linux distribution, download asterisk from its website and install it.
3. There are several softwares I forget when I made the appliance, it is not the mandatory (dependencies) but they are useful when we want to use asterisk optimally. They are: mpg123, sox, libmad, and festival. The easiest way to install it in openSUSE is using zypper. Check it first where they reside in repositories and add the repositories accordingly. mpg123 and sox are in the packman repository, libmad in OBS (please check with webpin) and festival in oss. Then as root run: "zypper install mpg123 sox libmad0 festival".
4. It is always useful to update your installation to update repository, to make sure that all the security update is up to date.
5. Download the latest dahdi from trunk and install it. Don't forget you should connected to internet to run this command as root

“svn co <http://svn.digium.com/svn/dahdi/linux/trunk> dahdi-linux”

“cd dahdi-linux”

“make”

and follow the instructions on the screen.

If all the installation successful, then you will have :

/etc/dahdi/

/etc/asterisk/

/var/lib/asterisk/

/var/spool/asterisk/

/etc/init.d/dahdi

/etc/init.d/asterisk161

/usr/sbin/asterisk

/usr/sbin/dahdi_genconf (and several dahdi-tools files)

Connect the telephone line(s) to your digium. Make sure that all the telephone lines are functioning before you connect it (please pay your bill if you don't to that yet, otherwise the announcement in the telephone lines will screw up your asterisk :-)).

As root run “/usr/sbin/dahdi_genconf”. This command will generate the automatic configuration for digium card in file /etc/dahdi/system.conf. In my server it contains:

```
# Autogenerated by /usr/sbin/dahdi_genconf on Tue Nov 17 18:38:30 2009
```

```
# If you edit this file and execute /usr/sbin/dahdi_genconf again,
```

```
# your manual changes will be LOST.
```

```
# Dahdi Configuration File
```

```
#
```

```
# This file is parsed by the Dahdi Configurator, dahdi_cfg
```

```
#
```

```
# Span 1: WCTDM/0 "Wildcard TDM410P Board 1" (MASTER)
```

```
fxsks=1
```

```
echocanceller=mg2,1
```

```
fxsks=2
```

```
echocanceller=mg2,2
```

```
fxsks=3
```

```
echocanceller=mg2,3
```

```
fxsks=4
```

```
echocanceller=mg2,4
```

```
# Span 2: WCTDM/1 "Wildcard TDM410P Board 2"
fxsks=5
echocanceller=mg2,5
fxsks=6
echocanceller=mg2,6
fxsks=7
echocanceller=mg2,7
fxsks=8
echocanceller=mg2,8
```

```
# Span 3: WCTDM/2 "Wildcard TDM410P Board 3"
fxsks=9
echocanceller=mg2,9
fxsks=10
echocanceller=mg2,10
fxsks=11
echocanceller=mg2,11
fxsks=12
echocanceller=mg2,12
```

```
# Global data
```

```
loadzone      = nl
defaultzone    = nl
```

Actually default loadzone and defaultzone is “us” but I change it to “nl” which is according to ITU is close to Indonesia signaling system. Please check ITU Operational Bulletin No. 781 – 1.II.2003. At least busy tone, congestion tone, and dial tone are running in the same frequency and cadence. If you want you can also rebuild asterisk so that it already contain the frequency and cadence for your country.

Besides /etc/dahdi/system.conf, dahdi_genconf will also automatically configure the file /etc/asterisk/dahdi-channels.conf. In my installation the content of the file is:

```
; Autogenerated by /usr/sbin/dahdi_genconf on Tue Nov 17 18:38:30 2009
; If you edit this file and execute /usr/sbin/dahdi_genconf again,
; your manual changes will be LOST.
; Dahdi Channels Configurations (chan_dahdi.conf)
;
; This is not intended to be a complete chan_dahdi.conf. Rather, it is intended
; to be #include-d by /etc/chan_dahdi.conf that will include the global settings
;
```

```
; Span 1: WCTDM/0 "Wildcard TDM410P Board 1" (MASTER)
;;; line="1 WCTDM/0/0"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn
channel => 1
callerid=
```

group=
context=default

;;; line="2 WCTDM/0/1"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn
channel => 2
callerid=
group=
context=default

;;; line="3 WCTDM/0/2"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn
channel => 3
callerid=
group=
context=default

;;; line="4 WCTDM/0/3"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn
channel => 4
callerid=
group=
context=default

; Span 2: WCTDM/1 "Wildcard TDM410P Board 2"
;;; line="5 WCTDM/1/0"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn
channel => 5
callerid=
group=
context=default

;;; line="6 WCTDM/1/1"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn

channel => 6
callerid=
group=
context=default

;;; line="7 WCTDM/1/2"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn
channel => 7
callerid=
group=
context=default

;;; line="8 WCTDM/1/3"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn
channel => 8
callerid=
group=
context=default

; Span 3: WCTDM/2 "Wildcard TDM410P Board 3"
;;; line="9 WCTDM/2/0"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn
channel => 9
callerid=
group=
context=default

;;; line="10 WCTDM/2/1"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn
channel => 10
callerid=
group=
context=default

;;; line="11 WCTDM/2/2"
signalling=fxs_ks
callerid=asreceived

```

group=0
context=from-pstn
channel => 11
callerid=
group=
context=default

;;; line="12 WCTDM/2/3"
signalling=fxs_ks
callerid=asreceived
group=0
context=from-pstn
channel => 12
callerid=
group=
context=default

```

The core of the asterisk configuration is dialplan. Dialplan manage how asterisk handle all the incoming and outgoing call. It can consist of 3 lines but also can reach tenth or hundreds lines, depends on how the complexity of our configuration. We can also use macro feature on asterisk. Dialplan is placed on /etc/asterisk/extensions.conf. My extensions.conf manage how the incoming call should be handled, how to make outgoing call to PSTN, GSM line and sip extensions, how to make conference call, how to connect to other asterisk server using IAX2 protocol, use the monitor application to record the conversation and how to make greeting. I will explain our extensions.conf in more detail in the next post together with sip.conf, iax.conf, meetme.conf and voicemail.conf.

PART 3

To enable asterisk to communicate with PSTN lines we should have either a VOIP-PSTN gateway or FXO card. I will not explain about VOIP-PSTN gateway, there are some service providers out there who provides this service for their customers. In my work I use Digium TDM 410P with 4 FXO port per card. There are some alternatives in the market like Sangoma, Rhino, etc, the important is we should make sure that it works with Asterisk either with dahdi driver or zaptel/zapata driver. Also if possible select the card that already has hardware echo-canceler. Echo is a problem in voip communication, and if you have card with no echo-canceler than your server CPU will busy do the job.

Just remember that Digium cards are no longer use zapata driver, and some changes has been made to the configuration file name and location, /etc/zaptel.conf become /etc/dahdi/system.conf and /etc/asterisk/zapata.conf become /etc/asterisk/chan_dahdi.conf

In the client site you can use any SIP client hardwares or softwares. Ekiga and Emphaty are the good choice for you who prefer GTK libraries and KCall and KPhone are for you who prefer Qt libraries. IP phone hardware now widely available in the market from cheap to high price, you can select any brand as long as it compatibles with Asterisk. In this project I choose Polycom IP-330, I also used Grandstream and Aastra in other implementation. In this implementation the owner also ask me to use Polycom KIRK Wireless Server 600V3 with Polycom DECT Handset 3040.

Now the time for the dialplan, extensions.conf, which is the core of asterisk implementation, as an example let me introduce you with my configuration. It is a good habit to always backup default asterisk configuration, and start the new configuration from the scratch.

My extensions.conf is:

```
; extensions.conf - the Asterisk dial plan
; Created by M. Edwin Z for xxxxxxxxxxxxxxxxx
; medwinz@gmail.com
;
; Static extension configuration file, used by
; the pbx_config module. This is where you configure all your
; inbound and outbound calls in Asterisk.
;
; This configuration file is reloaded
; - With the "dialplan reload" command in the CLI
; - With the "reload" command (that reloads everything) in the CLI
;
```

```
[general]
static=yes
writeprotect=yes
```

```
[globals]
RINGDELAY => 20
DYNAMIC_FEATURES => automon
```

```
[incoming]
exten => s,1,Answer
exten => s,2,Background(en/greeting-indonesia)
exten => s,3,Hangup()
```

```
exten => h,1,Hangup()
```

```
exten => 9999,1,VoiceMailMain()
exten => asterisk,1,VoicemailMain()
```

```
exten => 5000,1,Set(CHANNEL(language)=en) ; conference 1
exten => 5000,2,Meetme(5000)
exten => 5000,3,Hangup
```

```
exten => 6000,1,Set(CHANNEL(language)=en) ; conference 2
exten => 6000,2,Meetme(6000)
exten => 6000,3,Hangup
```

```
exten => 7000,1,Set(CHANNEL(language)=en) ; conference 3
exten => 7000,2,Meetme(7000)
exten => 7000,3,Hangup
```

```
exten => _XXXX,1,Answer
exten => _XXXX,2,Dial(SIP/${EXTEN},${RINGDELAY},t)
exten => _XXXX,3,Voicemail(${EXTEN}@default,u)
exten => _XXXX,4,Hangup()
```

```
exten => _XXXX,103,VoiceMail($ {EXTEN}@default,b)
exten => _XXXX,104,Hangup
```

```
[internal-fxo]
```

```
exten => s,1,Answer
exten => s,2,Wait(1)
exten => s,3,Background(en/autoattendant)
exten => s,4,WaitExten(2)
exten => 5000,1,MeetMe(5000)
exten => 6000,1,MeetMe(6000)
exten => 7000,1,MeetMe(7000)
exten => _XXXX,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN}-${STRFTIME($
{EPOCH},,%Y%m%d-%H%M%S)},m)
exten => _XXXX,2,Dial(SIP/${EXTEN},$
{RINGDELAY},t)
exten => _XXXX,3,VoiceMail(su${EXTEN})
exten => _XXXX,4,Hangup()
exten => _XXXX,103,VoiceMail(sb${EXTEN})
exten => _XXXX,104,Hangup()
```

```
exten => h,1,Hangup()
```

```
exten => t,1,Monitor(wav,Call-${CALLERID(num)}-9019-${STRFTIME($ {EPOCH},,
%Y%m%d-%H%M%S)},m)
exten => t,2,Dial(SIP/9019&SIP/9006&SIP/9007&SIP/9001&SIP/9002&SIP/9015,$
{RINGDELAY},t)
exten => t,3,Hangup
exten =>
t,305,Dial(SIP/9001&SIP/9002&SIP/9003&SIP/9004&SIP/9005&SIP/9006&SIP/9007&SI
P/9008&SIP/9009&SIP/9010&SIP/9011&SIP/9016&SIP/9017&SIP/9018&SIP/9019)
```

```
exten => t,306,Hangup
```

```
include => incoming
```

```
[internal-fxs]
```

```
include => incoming
```

```
[internal-sip]
```

```
exten => _1.,1,Dial(IAX2/ygpabxsv:0000@10.1.1.120/${EXTEN:1}@local)
exten => _1.,2,Hangup()
```

```
exten => _2.,1,Dial(IAX2/ygpabxsv:0000@10.7.1.120/${EXTEN:1}@local)
exten => _2.,2,Hangup()
```

```
::GSM call to Telkomsel/HALO
```

```
exten => _000811.,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN:2}-${
{STRFTIME($ {EPOCH},,%Y%m%d-%H%M%S)},m)
```

exten => _000811.,2,Dial(SIP/9031/\${EXTEN:1})

exten => _000811.,3,Hangup

::GSM call to Telkomsel/Simpati

exten => _000812.,1,Monitor(wav,Call-\${CALLERID(num)}-\${EXTEN:2}-\${STRFTIME(\${EPOCH}.,%Y%m%d-%H%M%S)},m)

exten => _000812.,2,Dial(SIP/9031/\${EXTEN:1})

exten => _000812.,3,Hangup

::GSM call to Telkomsel/Simpati

exten => _000813.,1,Monitor(wav,Call-\${CALLERID(num)}-\${EXTEN:2}-\${STRFTIME(\${EPOCH}.,%Y%m%d-%H%M%S)},m)

exten => _000813.,2,Dial(SIP/9031/\${EXTEN:1})

exten => _000813.,3,Hangup

::GSM call to Telkomsel/As

exten => _000852.,1,Monitor(wav,Call-\${CALLERID(num)}-\${EXTEN:2}-\${STRFTIME(\${EPOCH}.,%Y%m%d-%H%M%S)},m)

exten => _000852.,2,Dial(SIP/9031/\${EXTEN:1})

exten => _000852.,3,Hangup

::GSM call to Telkomsel/As

exten => _000853.,1,Monitor(wav,Call-\${CALLERID(num)}-\${EXTEN:2}-\${STRFTIME(\${EPOCH}.,%Y%m%d-%H%M%S)},m)

exten => _000853.,2,Dial(SIP/9031/\${EXTEN:1})

exten => _000853.,3,Hangup

::GSM call to Indosat

exten => _000814.,1,Monitor(wav,Call-\${CALLERID(num)}-\${EXTEN:2}-\${STRFTIME(\${EPOCH}.,%Y%m%d-%H%M%S)},m)

exten => _000814.,2,Dial(SIP/9031/\${EXTEN:1})

exten => _000814.,3,Hangup

::GSM call to Indosat

exten => _000815.,1,Monitor(wav,Call-\${CALLERID(num)}-\${EXTEN:2}-\${STRFTIME(\${EPOCH}.,%Y%m%d-%H%M%S)},m)

exten => _000815.,2,Dial(SIP/9031/\${EXTEN:1})

exten => _000815.,3,Hangup

::GSM call to Indosat

exten => _000816.,1,Monitor(wav,Call-\${CALLERID(num)}-\${EXTEN:2}-\${STRFTIME(\${EPOCH}.,%Y%m%d-%H%M%S)},m)

exten => _000816.,2,Dial(SIP/9031/\${EXTEN:1})

exten => _000816.,3,Hangup

::GSM call to Indosat

exten => _000855.,1,Monitor(wav,Call-\${CALLERID(num)}-\${EXTEN:2}-\${

```
{STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)},m)
exten => _000855.,2,Dial(SIP/9031/${EXTEN:1})
exten => _000855.,3,Hangup
```

::GSM call to Indosat

```
exten => _000856.,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN:2}-${
{STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)},m)
exten => _000856.,2,Dial(SIP/9031/${EXTEN:1})
exten => _000856.,3,Hangup
```

::GSM call to Indosat

```
exten => _000857.,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN:2}-${
{STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)},m)
exten => _000857.,2,Dial(SIP/9031/${EXTEN:1})
exten => _000857.,3,Hangup
```

::GSM call to Indosat

```
exten => _000858.,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN:2}-${
{STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)},m)
exten => _000858.,2,Dial(SIP/9031/${EXTEN:1})
exten => _000858.,3,Hangup
```

::GSM call to XL

```
exten => _000817.,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN:2}-${
{STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)},m)
exten => _000817.,2,Dial(SIP/9032/${EXTEN:1})
exten => _000817.,3,Hangup
```

::GSM call to XL

```
exten => _000818.,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN:2}-${
{STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)},m)
exten => _000818.,2,Dial(SIP/9032/${EXTEN:1})
exten => _000818.,3,Hangup
```

::GSM call to XL

```
exten => _000819.,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN:2}-${
{STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)},m)
exten => _000819.,2,Dial(SIP/9032/${EXTEN:1})
exten => _000819.,3,Hangup
```

::GSM call to XL

```
exten => _000859.,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN:2}-${
{STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)},m)
exten => _000859.,2,Dial(SIP/9032/${EXTEN:1})
exten => _000859.,3,Hangup
```

::GSM call to XL

```
exten => _000878.,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN:2}-${
```

```
{STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)},m)
exten => _000878.,2,Dial(SIP/9032/${EXTEN:1})
exten => _000878.,3,Hangup
```

```
;GSM call to 3
```

```
exten => _000898.,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN:2}-${
{STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)},m)
exten => _000898.,2,Dial(SIP/9032/${EXTEN:1})
exten => _000898.,3,Hangup
```

```
;GSM call to 3
```

```
exten => _000899.,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN:2}-${
{STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)},m)
exten => _000899.,2,Dial(SIP/9032/${EXTEN:1})
exten => _000899.,3,Hangup
```

```
::GSM call to Axis
```

```
; exten => _000831.,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN:2}-${
{STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)},m)
; exten => _000831.,2,Dial(SIP/9032/${EXTEN:1})
; exten => _000831.,3,Hangup
```

```
::GSM call to Axis
```

```
; exten => _000838.,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN:2}-${
{STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)},m)
;exten => _000838.,2,Dial(SIP/9032/${EXTEN:1})
;exten => _000838.,3,Hangup
```

```
include => global
```

```
include => incoming
```

```
[global]
```

```
exten => _0.,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN:1}-${STRFTIME($
{EPOCH},,%Y%m%d-%H%M%S)},m)
exten => _0.,2,Dial(DAHDI/g1/${EXTEN:1})
exten => _0.,3,Hangup
exten => _0.,103,Playback(en/tt-allbusy)
exten => _0.,104,Hangup
```

```
[recordings]
```

```
exten => 500,1,Answer
exten => 500,2,Playback(en/silakanrekamgreeting)
exten => 500,3,Record(en/mymessage:gsm)
exten => 500,4,Playback(en/pesananda)
exten => 500,5,Playback(en/mymessage)
exten => 500,6,Playback(en/tekan1)
exten => 500,7,WaitExten(3)
```

```

exten => t,1,Playback(en/maafmohonulangi)
exten => t,2,Goto(500,5)

exten => i,1,Playback(en/pesanandasalah)
exten => i,2,Goto(500,5)

exten => 1,1,System(/bin/mv /var/lib/asterisk/sounds/en/mymessage.gsm
/var/lib/asterisk/sounds/en/autoattendant.gsm)
exten => 1,2,Playback(en/terimakasih)
exten => 1,3,Playback(en/tekan3)

exten => 2,1,Goto(500,1)
exten => 3,1,Goto(500,1)
exten => 4,1,Hangup

include => internal-sip

[local]
;
; Master context for local, toll-free, and iaxtel calls only
;
ignorepat => 9
include => default
include => parkedcalls
include => internal-sip

[default]
include => internal-sip

```

PART 4

Pada part 3 saya telah memberikan contoh extensions.conf, saya perlu menyertakan beberapa contoh file konfigurasi lain yang dibutuhkan agar penjelasan extension.conf bisa dimengerti. File-file tersebut adalah:

1. /etc/asterisk/chan_dahdi.conf
2. /etc/asterisk/sip.conf
3. /etc/asterisk/iax.conf
4. /etc/asterisk/meetme.conf
5. /etc/asterisk/voicemail.conf

Contoh chan_dahdi.conf:

```

;
; dahdi_channels.conf configuration of digium card
;
; Configuration file

[channels]

```

```
language=en
context=internal-fxo
signalling=fxs_ks
rxwink=300
cidstart=polarity ; jangan ada line yang ngutang akan mengacaukan DTMF dan cid
signalling
```

```
answeronpolarityswitch=no
hanguponpolarityswitch=no
;cidstart=ring ; ini test saja
pulsedial=no
;useincomingcalleridondahditransfer=yes
cidsignalling=dtmf
busydetect=yes
busycount=6
```

```
usecallerid=yes
callerid=asreceived
hidecallerid=no
callwaiting=yes
usecallingpres=yes
callwaitingcallerid=yes
threewaycalling=yes
transfer=yes
cancallforward=yes
callreturn=yes
echocancel=yes
echocancelwhenbridged=no
echotraining=800
rxgain=3.0
txgain=0.0
group=0
callgroup=1
pickupgroup=1
immediate=no
faxdetect=no
group=1
disallow=all
allow=all
```

```
echocanceller=mg2,1-12
channel => 1-12
```

Contoh sip.conf:

```
[general]
port = 5060
bindaddr = 10.8.1.120
disallow=all
```

allow=all
allow=ulaw
allow=gsm
context=internal-sip

[9001]
type=friend
host=dynamic
dtmfmode=rfc2833
language = en
context=recordings
nat=no
username=YGTELEPH01
userid=9001
callerid=YGTELEPH01 <9001>
mailbox=9001
allow=all
qualify=yes

[9002]
type=friend
host=dynamic
dtmfmode=rfc2833
language = en
context=internal-sip
nat=no
username=YGTELEPH02
userid=9002
callerid=YGTELEPH02 <9002>
mailbox=9002
allow=all
qualify=yes

;tambahkan sesuai extension yang anda miliki

[9031]
type=peer
insecure=very
disallow=all
allow=ulaw
allow=alaw
allow=gsm
context=internal-sip
host=10.8.1.31
username=GS8
permit=10.8.1.31/255.255.255.255
qualify=yes
canreinvite=no
call-limit=4

```
dtmfmode=rfc2833
nat=no

[9032]
type=peer
insecure=very
disallow=all
allow=ulaw
allow=alaw
allow=gsm
context=internal-sip
host=10.8.1.32
username=GS8
permit=10.8.1.32/55.255.255.255
qualify=yes
canreinvite=no
call-limit=4
dtmfmode=rfc2833
nat=no
```

Contoh iax.conf:

```
; Inter-Asterisk eXchange driver definition
;
; This configuration is re-read at reload
; or with the CLI command
;   reload chan_iax2.so
;
; General settings, like port number to bind to, and
; an option address (the default is to bind to all
; local addresses).
;
[general]
bindport=4569
bindaddr=10.8.1.120
delayreject=yes
language=en
bandwidth=high
disallow=all
allow=ulaw
allow=alaw
allow=gsm
jittertargetextra=40
jitterbuffer=yes
dropcount=3
maxjitterbuffer=300
minjitterbuffer=300
minexcessbuffer=200
mailboxdetail=yes
```

autokill=yes

register => ncpabxsv:0000@10.1.1.120:4569

register => dppabxsv:0000@10.7.1.120:4569

register => jbpabxsv:0000@10.9.1.120:4569

tos=0x10

[guest]

type=user

context=default

callerid="Guest IAX User"

;

; Trust Caller*ID Coming from iaxtel.com

;

[iaxtel]

type=user

context=default

auth=rsa

inkeys=iaxtel

;

; Trust Caller*ID Coming from iax.fwdnet.net

;

[iaxfwd]

type=user

context=default

auth=rsa

inkeys=freeworlddialup

[ncpabxsv]

type=friend

auth=md5

secret=0000

context=local

host=dynamic

defaultip=10.1.1.120

qualify=yes

requirecalltoken=no

[dppabxsv]

type=friend

auth=md5

secret=0000

context=local

host=dynamic

defaultip=10.7.1.120

```
qualify=yes
requirecalltoken=no
```

```
[ygpabxsv]
type=friend
auth=md5
secret=0000
context=local
host=dynamic
defaultip=10.8.1.120
qualify=yes
requirecalltoken=no
```

```
[jbabxsv]
type=friend
auth=md5
secret=0000
context=local
host=dynamic
defaultip=10.9.1.120
qualify=yes
requirecalltoken=no
```

Contoh meetme.conf:

```
[rooms]
;#include meetme_additional.conf

conf => 5000
conf => 6000
conf => 7000
```

Contoh voicemail.conf:

```
;
; Voicemail Configuration
;

;
; NOTE: Asterisk has to edit this file to change a user's password. This does
; not currently work with the "#include <file>" directive for Asterisk
; configuration files, nor when using realtime static configuration.
; Do not use them with this configuration file.
;

[general]
format=wav
serveremail=asterisk
fromstring=Asterisk PABX
sendvoicemail=yes
```

language=en
operator=no
envelope=yes
attach=yes
maxmsg=20
maxsecs=180
minsecs=6
maxgreet=60
skipms=3000
maxsilence=5
silencethreshold=128
maxlogins=3

emailbody=Anda mempunyai pesan baru

emaildateformat=%A, %d %B %Y at %H:%M:%S
mailcmd=/usr/sbin/sendmail -t

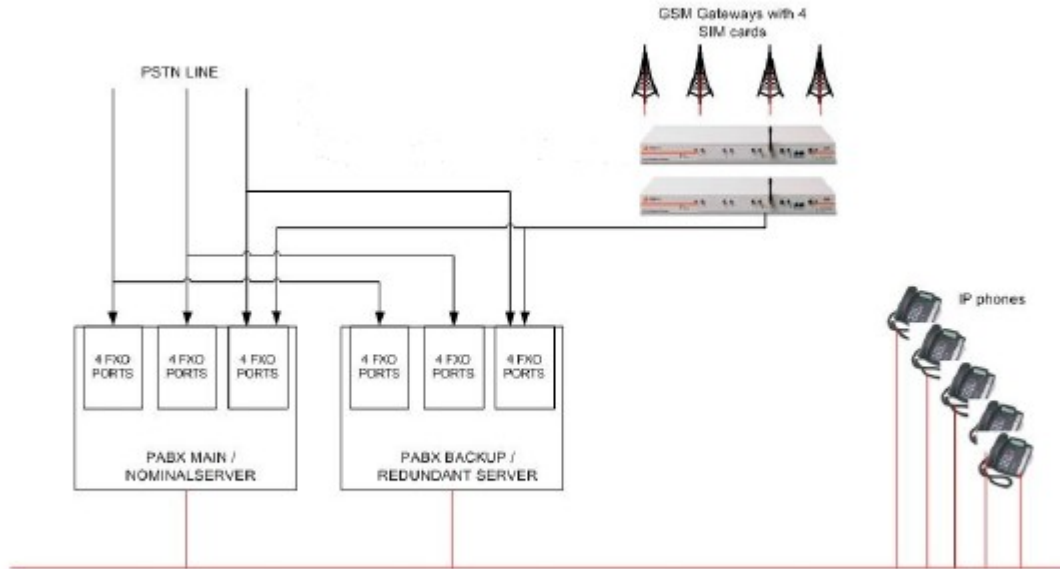
sendvoicemail=yes ; Allow the user to compose and send a voicemail while inside
; VoiceMailMain() [option 5 from mailbox's advanced menu].
; If set to 'no', option 5 will not be listed.

[default]
; isikan sebanyak extension yang anda miliki

9001 => 9001,medwinz,,,attach=no
9002 => 9002,medwinz,,,attach=no

[zonemessages]
yogyakarta=Asia/Jakarta|'vm-received' Q 'digits/at' R

Desain yang saya buat ini secara sederhana digambarkan dalam diagram dibawah



Ip phones mempunyai extension 9001 sampai dengan 9027. GSM gateway diperlakukan sebagai sip extension dengan nomer extension 9031 dan 9032. Lihat file sip.conf. Bagaimana membuat agar sebuah ip phone mempunyai nomor ekstensi? Ini tergantung dari ip phone yang anda gunakan, untuk langkah awal anda dapat menset sebuah dhcp untuk kemudian setiap ip-phone akan mengambil sebuah ip. Biasanya didalam sebuah ip-phone sudah ditanam sebuah webserver yang dapat diakses dari browser untuk selanjutnya kita beri nomor ekstensi. Proses ini sering dinamakan provision. Favorit saya untuk mem-provisi ip-phone adalah dengan menset sebuah ftp server yang kemudian setiap ip-phone akan mendownload konfigurasi dari ftp server tersebut. Lebih lengkapnya silakan baca manual ip-phone anda.

Sekarang mari kita mulai membahas file extensions.conf. Yang penting diketahui bahwa dialplan itu terdiri dari beberapa context. Context ditandai dengan [...], misalnya [incoming], [internal-fxo], [internal-sip] dsb. Context ini saling berhubungan antara extensions.conf dengan file-file yang lain. Context akan mengatur perlakuan terhadap suatu incoming atau outgoing call oleh asterisk. Asterisk mengenal beberapa standard ekstensi yaitu:

- i : invalid
- s : start
- h : hangup
- t : timeout
- T : absolute timeout
- a : asterisk extension
- o : operator

Yang biasanya sering digunakan adalah:

- s : start : apa yang harus dilakukan oleh asterisk kalau ada incoming call
- i : invalid entry : apa yang dilakukan kalau entry yang dimasukkan salah

- t : time out : apa yang dilakukan kalau timeout sudah lewat

Sekarang coba kita perhatikan syntax extensions.conf berikut:

```
[internal-fxo]
exten => s,1,Answer
exten => s,2,Wait(1)
exten => s,3,Background(en/autoattendant)
exten => s,4,WaitExten(2)
```

Maka artinya kurang lebih adalah untuk context internal-fxo kalau ada telepon yang masuk maka yang harus dilakukan oleh asterisk adalah:

1. dijawab (diangkat) -> s,1,Answer
2. tunggu selama 1 detik -> s,2,Wait(1)
3. jalankan di latar belakang file /var/lib/asterisk/sound/en/autoattendant.gsm -> s,3,Background(en/autoattendant)
4. tunggu input keypad selama 2 detik -> s,4,WaitExten(2)

Mudahkan?

Asterisk mempunyai beberapa aplikasi yang bisa dipanggil melalui extensions.conf yang saya gunakan di sini adalah VoiceMail yaitu aplikasi untuk meninggalkan pesan jika telepon tidak diangkat atau sibuk, meetme untuk melakukan *conference call* (percakapan dengan peserta lebih dari 2 orang), dan Monitor untuk merekam suatu percakapan ke dalam file. Mari kita lihat contoh extensions.conf:

```
[internal-fxo]
.....

exten => 5000,1,MeetMe(5000)
exten => 6000,1,MeetMe(6000)
exten => 7000,1,MeetMe(7000)
```

Perhatikan juga contoh meetme.conf:

```
[rooms]
;#include meetme_additional.conf

conf => 5000
conf => 6000
conf => 7000
```

Kita telah mendefinisikan 3 ruangan untuk melakukan konferensi yaitu ekstensi 5000, 6000, dan 7000. Kemudian pada context [internal-fxo] di extensions.conf kita definisikan bahwa user yang mengakses ekstensi 5000, 6000 dan 7000 akan masuk ke ruangan konferensi. Mudahkan. Perlu diketahui bahwa ruangan konferensi ini tidak hanya bisa diakses oleh ekstensi lokal tetapi juga dari telepon di tempat lain, baik voip, GSM, atau PSTN. Misalnya kita ingin mengajak rekan kita yang kebetulan sedang diluar kantor untuk ikut meeting, maka kita dapat menghubungi handphonenya dan selanjutnya kita transfer ke 5000, 6000 atau 7000.

VoiceMail cukup mudah untuk dikonfigurasi jika kita menginginkannya. Ada beberapa flag yang digunakan untuk mengatur VoiceMail yaitu:

- s : jika diberikan akan membuat pesan "Please leave your message after the tone. When done, hang up, or press the pound key" tidak dimainkan
- u: jika diberikan akan memutar pesan "The person at extension ... 1234 ... is unavailable"
- b: jika diberikan akan memutar pesan "The person at extension ... 1234 ... is busy"

Kita dapat mengkombinasikan flag tersebut misalnya:

- su : pesan unavailable akan diputar tetapi pesan instruksi tidak
- sb : pesan busy akan diputar tetapi pesan instruksi tidak
- u : pesan unavailable akan diputar dilanjutkan dengan pesan instruksi
- b : pesan busy akan diputar dilanjutkan dengan pesan instruksi

Pada context [incoming] di extensions.conf saya mendefinisikan:

```
exten => _XXXX,1,Answer
exten => _XXXX,2,Dial(SIP/${EXTEN},${RINGDELAY},t)
exten => _XXXX,3,VoiceMail(${EXTEN}@default,u)
exten => _XXXX,4,Hangup()
exten => _XXXX,103,VoiceMail(${EXTEN}@default,b)
exten => _XXXX,104,Hangup
```

Maksudnya kurang lebih adalah kalau ada yang men-dial ekstensi XXXX (sesuai dengan yg telah didefinisikan di sip.conf) misalnya 9001 maka:

1. jawab
2. dial extension selama 20 detik (ini ditentukan pada context [globals] RINGDELAY => 20), kalau sudah lewat 20 detik maka
3. putar pesan unavailable dilanjutkan dengan instruksi untuk menyimpan pesan.
4. hangup
5. jika nada sibuk, putar pesan sibuk dilanjutkan dengan instruksi untuk menyimpan pesan
6. hangup

Pesan yang masuk akan disimpan pada /var/spool/asterisk/voicemail/context/boxnumber/INBOX. Misalnya dalam kasus di atas maka :

- context = default, sesuai exten => _XXXX,3,VoiceMail(\${EXTEN}@default,u)
- boxnumber adalah mailbox untuk nomer ekstensi tertentu, misalnya untuk ekstensi 9001 kebetulan saya set mailbox=9001 sama dengan nomor ekstensinya. Lihat file sip.conf di atas.
- maka jika penelpon menelpon 9001 dan meninggalkan voicemail maka lokasi penyimpanannya pada /var/spool/asterisk/voicemail/default/9001/INBOX

Kita bisa menyimpan semua percakapan yang terjadi melalui asterisk dengan memanfaatkan aplikasi Monitor. Tentu saja untuk mengkonfigurasinya anda harus menanyakan policy mengenai hal ini kepada pemilik jaringan/asterisk di mana anda memasangnya. Karena hal ini berhubungan dengan privacy. Ada beberapa hal yang sebaiknya diperhatikan dalam mensetup Monitor, standar syntax adalah sebagai berikut: Monitor(ext,basename,flags). Penjelasan sederhananya adalah sebagai berikut:

- ext : format sound file, defaultnya adalah .wav
- basename : dalam contoh saya menggunakan Call-\${CALLERID(num)}-\${EXTEN}-\${STRFTIME(\${EPOCH},,%Y%m%d-%H%M%S, ini akan mengakibatkan file disimpan dengan nama misalnya Call-8001-02125558785-20091222-161031.wav dimana 8001 adalah nomer ekstensi, 02125558785 adalah nomer yang dituju, 20091222 adalah tanggal-bulan-tahun, 161031 adalah jam-menit-detik. Demikian juga kalau incoming call, kalau anda berlangganan

CID maka asterisk bisa membaca asterisk yang masuk, tetapi sekiranya anda tidak berlangganan CID maka incoming call akan disimpan dengan nama misalnya Call- -8019-20091222-122545.wav

- m : adalah flag yang bila digunakan maka asterisk akan memanggil program diluar asterisk untuk mengkombinasikan dua buah sound file, in dan out, ke dalam sebuah file. Program yang dipanggil adalah sox. Kadang-kadang sox tidak bisa mengenali dan menggabungkan format sound (alaw) akibatnya seringkali kita menemukan untuk sebuah percakapan masih terdapat dua buah file, in dan out. Misalnya : Call-8019-723964-20091222-151827-in.wav dan Call-8019-723964-20091222-151827-out.wav

Untuk mengaktifkan Monitor tidaklah sulit, sebagai contoh perhatikan lagi file extensions.conf:

```
[internal-fxo]
```

```
.....
```

```
exten => _XXXX,1,Monitor(wav,Call-${CALLERID(num)}-${EXTEN}-${STRFTIME($
{EPOCH},,%Y%m%d-%H%M%S)},m)
exten => _XXXX,2,Dial(SIP/${EXTEN},$
{RINGDELAY},t)
exten => _XXXX,3,Voicemail(su${EXTEN})
exten => _XXXX,4,Hangup()
exten => _XXXX,103,Voicemail(sb${EXTEN})
exten => _XXXX,104,Hangup()
```

Maksud dari baris ini:

1. untuk extensi xxxx, rekam percakapan dengan format Call-no.extensi-tanggal-jam
2. dial extensi xxxx dan dering selama 20 detik (masih ingat ya, yang diatas)
3. kalau lewat 20 detik maka aktifkan voicemail
4. kala nada sibuk aktifkan voicemail
5. hangup

Hmm... banyak juga ya. Mudah-mudahan tidak memusingkan. Masih ada beberapa hal di dalam extensions.conf yang akan saya jelaskan misalnya bagaimana mengkoneksi asterisk server di lokasi lain, bagaimana merekam pesan (recording untuk greeting), dan terutama pengaturan context yang berkaitan dengan channel dahdi dan sip.conf.

PART 5

I will explain a bit more deeper about Asterisk configuration in this post, some trick and useful configuration that I found really helpful in configuring asterisk instalation. Asterisk developer really did a good job to make a complete PBX, they give the best tools to us and now it is our job to configure it.

One thing I found really annoying is the echo if we connect asterisk to PSTN line. I use digium TDM 410P and leave the card without tune it will give annoying echo. In my [earlier post](#) I explain that by running /usr/sbin/dahdi_genconf dahdi will automatically create /etc/dahdi/system.conf file that already contain information about hardware echo canceller. First thing you should remember if you have the budget is buy a card with hardware echo canceller. It will let the card to manage the echo without give the processor too much task to reduce it. After that you should tune the card. Luckily Digium give the

best tools to tune the card named fxotune. To tune your card first shutdown the asterisk service and then run:

```
# /usr/sbin/fxotune -i 0
```

I put 0 (zero) because to dial an outside line I set the asterisk configuration to use 0. You should change it to whatever number you use. fxotune will create /etc/fxotune.conf file to put all the configuration it creates to reduce the echo. Please read "man fxotune" for more explanation about the tools.

To enable Asterisk to use the card configuration every time we boot the server we need a slight modification of init script. We should call the fxotune before we call Asterisk, you can either modify the init script of Asterisk to call fxotune before it call the Asterisk or you can modify /etc/init.d/after.local (yes, I use openSUSE). I prefer to use after.local. Create /etc/init.d/after.local and fill the lines below:

```
# ! /bin/sh
```

```
/usr/sbin/fxotune -s
```

```
sleep 1
```

```
/etc/init.d/asterisk161 start
```

Also you need to remove asterisk service from init script

```
# insserv -r /etc/init.d/asterisk161
```

You can boot the server and hear the difference. If everything goes well you can hear no echo

Now let's take a look back to our extension.conf that I already give in [the part 3](#).

```
[recordings]
```

```
exten => 500,1,Answer
```

```
exten => 500,2,Playback(en/silakanrekamgreeting)
```

```
exten => 500,3,Record(en/mymessage:gsm)
```

```
exten => 500,4,Playback(en/pesananda)
```

```
exten => 500,5,Playback(en/mymessage)
```

```
exten => 500,6,Playback(en/tekan1)
```

```
exten => 500,7,WaitExten(3)
```

```
exten => t,1,Playback(en/maafmohonulangi)
```

```
exten => t,2,Goto(500,5)
```

```
exten => i,1,Playback(en/pesanandasalah)
```

exten => i,2,Goto(500,5)

exten => 1,1,System(/bin/mv /var/lib/asterisk/sounds/en/mymessage.gsm
/var/lib/asterisk/sounds/en/autoattendant.gsm)

exten => 1,2,Playback(en/terimakasih)

exten => 1,3,Playback(en/tekan3)

exten => 2,1,Goto(500,1)

exten => 3,1,Goto(500,1)

exten => 4,1,Hangup

include => internal-sip

This is the context about recording that we put it in extensions.conf and also in sip.conf, that's why I put "include => internal-sip" in the bottom of the context. We create the special extension 500 to record the greeting for our system. We will put the record as greeting, when someone from outside call our lines, then Asterisk will play this greeting. I will explain it to you don't worry . Those lines means:

1. If dial 500 then answer.
2. Play the sound file /var/lib/asterisk/sounds/en/silakanrekamgreeting.gsm. You can record a custom sound file which contain something like "Please record your greeting after the beep", save it as gsm format and call it from here.
3. Record your message (say the greeting you want to record) and put it as /var/lib/asterisk/sounds/en/mymessage.gsm
4. Play the file /var/lib/asterisk/sounds/en/pesananda.gsm. You can record a custom sound file which contain something like "Your greeting is", save it as gsm format and call it from here.
5. Play your record greeting that just you record in step 3.
6. Play the sound file /var/lib/asterisk/sounds/en/tekan1.gsm. You can record a custom sound file which contain something like "Please press 1 to save your message", save it as gsm format and call it from here.
7. Wait 3 second for pressing 1
8. t,1 means that if 3 seconds already time-out then play the sound file /var/lib/asterisk/sounds/en/maafmohoulangi.gsm. You can record a custom sound file which contain something like "Please re-record your message", save it as gsm format and call it from here.
9. t,2 go to point 6 above and repeat the steps.
- 10.i,1 means if you press another number in step 7 (you don't record the message) then play the sound file /var/lib/asterisk/sounds/en/pesanandasalah.gsm. You can record a custom sound file which contain something like "Sorry I didn't get that", save it as gsm format and call it from here.
- 11.i,2 go to point 6 above and repeat the steps.
- 12.1,1 if you press 1 in step 7 then asterisk will move /bin/mv /var/lib/asterisk/sounds/en/mymessage.gsm to /var/lib/asterisk/sounds/en/autoattendant.gsm
- 13.1,2 means play the sound file /var/lib/asterisk/sounds/en/terimakasih.gsm. You can record a custom sound file which contain something like "thankyou", save it as gsm format and call it

from here.

14.1,3 means play the sound file /var/lib/asterisk/sounds/en/tekan3.gsm. You can record a custom sound file which contain something like "press 3 to record another message", save it as gsm format and call it from here.

15.include => internal-sip, means that asterisk will see also [recording] context in [internal-sip] context, usually we manage [internal-sip] in sip.conf.

Now it's time to configure /etc/asterisk/sip.conf. This file manage the sip for ip phone and other peripheral in our setup. I use several desk ip-phone, wireless ip-phone (Polycom Kirk DECT) and also GSM gateway (Zed, Musitel etc), all of it running well. My sip.conf looks something like:

```
[general]
```

```
port = 5060
```

```
bindaddr = 10.7.1.120
```

```
disallow=all
```

```
allow=all
```

```
allow=ulaw
```

```
allow=gsm
```

```
context=internal-sip
```

```
;—————xxxxx site—————
```

```
;Polycom IP330
```

```
[8001]
```

```
type=friend
```

```
host=dynamic
```

```
dtmfmode=rfc2833
```

```
language=en
```

```
context=internal-sip
```

```
nat=no
```

```
canreinvite=no
```

```
username=TELEPH01
```

userid=8001

callerid=TELEPH01 <8001>

mailbox=8001

allow=all

qualify=yes

[8002]

type=friend

host=dynamic

dtmfmode=rfc2833

language=en

context=internal-sip

nat=no

username=TELEPH02

userid=8002

callerid=TELEPH02 <8002>

mailbox=8002

allow=all

qualify=yes

[8006]

type=friend

host=dynamic

dtmfmode=rfc2833

language=en

;context=internal-sip

context=recordings

nat=no

canreinvite=no

username=TELEPH06

userid=8006

callerid=DPTELEPH06 <8006>

mailbox=8006

allow=all

qualify=yes

; KIRK DECT 3040 at site

[8020]

type=friend

host=dynamic

dtmfmode=rfc2833

language=en

context=internal-sip

nat=no

username=TELEPH20

userid=8020

callerid=TELEPH20 <8020>

mailbox=8020

allow=all

qualify=yes

;Zed GSM GATEWAY

[8031]

type=peer

insecure=very

disallow=all

allow=ulaw

allow=alaw

allow=gsm

context=internal-sip

host=10.7.1.31

username=GS8

permit=10.7.1.31/255.255.255.255

qualify=yes

canreinvite=no

call-limit=4

dtmfmode=rfc2833

nat=no

[8032]

type=peer

insecure=very

disallow=all

allow=ulaw

allow=alaw

allow=gsm

context=internal-sip

host=10.7.1.32

username=GS8

permit=10.7.1.32/255.255.255.255

qualify=yes

canreinvite=no

call-limit=4

dtmfmode=rfc2833

nat=no

[8001] and [8006] are the desk ip-phone, [8020] is wireless ip-phone, and [8031] and [8032] are Zed gsm gateway. Please pay attention to [8001] and [8006] almost all the configuration are same, except one line.context. For 8001 context=internal-sip but 8006 context=recording. This line tell us that for recording / greeting purpose we will use the phone with extension 8006. So we can only dial 500 and do the recording in that phone as we set it in /etc/asterisk/extension.conf. You cannot dial 500 and do recording in the phone with the sip configuration doesn't include the line context=recording. I hope you get it.

For gsm gateway, I set it up just like the other as internal-sip. But there are a lot of gsm gateways out there and every brand has their own configuration so please read the manual of your gsm gateway and set it up properly. You can use it in front of digium as fxo/fxs or you can also set it up as sip extension depend on you gsm gateway type.

Let's move to chan_dahdi.conf that I already attached several post earlier. Let me remind part of the file:

[channels]

language=en

context=internal-fxo

signalling=fxs_ks

rxwink=300

cidstart=polarity

answeronpolarityswitch=no

hanguponpolarityswitch=no

pulsedial=no

```
cidsignalling=dtmf
```

```
busydetect=yes
```

```
busycount=6
```

```
.....
```

```
echocanceller=mg2,1-12
```

```
channel => 1-12
```

there is line with "context=internal-fxo". Basically it means that all the channels 1 through 12 are in the context of internal-fxo. By doing this all the lines will follow the setup we already done in extensions.conf under context [internal-fxo], please take a look extensions.conf in earlier part. You got it, don't you ?

Let me finish this asterisk session, 5 part seem not enough but I'm afraid this will make my blog so bore. So final notes is about iax.conf. If you have several locations with asterisk server in every location you can connect the server and make conversation like you dial an extension. First of all you should setup a VPN between the site. I presume you already now how to set it up, there are a lot of howto in internet. Then you should configure the iax.conf. I already give the example in previous post, let me explain. Take attention in this part:

```
[general]
```

```
bindport=4569
```

```
bindaddr=10.8.1.120
```

```
.....
```

```
register => ncpabxsv:0000@10.1.1.120:4569
```

```
register => dppabxsv:0000@10.7.1.120:4569
```

```
register => ygpabxsv:0000@10.8.1.120:4569
```

This server address is 10.8.1.120 and the port 4569 use for the iax. We should register all the server including this server in the 3 lines at the bottom.

Also we should make the setup for every server like below:

```
[ncpabxsv]
```

type=friend

auth=md5

secret=0000

context=local

host=dynamic

defaultip=10.1.1.120

qualify=yes

requirecalltoken=no

[dppabxsv]

type=friend

auth=md5

secret=0000

context=local

host=dynamic

defaultip=10.7.1.120

qualify=yes

requirecalltoken=no

[ygpabxsv]

type=friend

auth=md5

secret=0000

context=local

host=dynamic

defaultip=10.8.1.120

qualify=yes

requirecalltoken=no

In all site with the asterisk server we should configure iax.conf so every server can be registered with each other. By doing this you can call other site with extension. Now take a look again our extensions.conf in this section:

[internal-sip]

exten => _1.,1,Dial(IAX2/ygpabxsv:0000@10.1.1.120/\${EXTEN:1}@local)

exten => _1.,2,Hangup()

exten => _2.,1,Dial(IAX2/ygpabxsv:0000@10.7.1.120/\${EXTEN:1}@local)

exten => _2.,2,Hangup()

Above lines means:

1. _1. -> if you start a call with "pressing 1 then follow by extension then you connect to asterisk server in ip address 10.1.1.120". 10.1.1.120/\${EXTEN:1} means the asterisk server in ip 10.1.1.120 will stripe the first digit.
2. _2. -> if you start a call with "pressing 2 then follow by extension then you connect to asterisk server in ip address 10.7.1.120". 10.7.1.120/\${EXTEN:1} means the asterisk server in ip 10.7.1.120 will stripe the first digit.

I think I already explain everything that you should know about how to setup and configure an asterisk server, make a conference room, setting up extension, and even connecting between two or more asterisk server using iax. Now it is your turn. You can experiment with my setting until you get used to it and try another configuration that match with what you want.

Don't forget to have a lot of fun.